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VoIP Analysis Performance of Quality of Service in Converged Networks

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ABSTRACT

One of the most common applications in use on converged networks today is Voice over Internet Protocol or VoIP.

VoIP allows for much more flexibility in handling multiple calls simultaneously and also helps companies save money by providing additional telephony services like three-way calling and call-forwarding without the company having to pay for them individually. The most key requirement for successful VoIP communications is a quality of service (QoS). Voice communications require networks with very low call delay, low jitter, higher MOS-LQ and minimal packet loss.

This thesis briefly describes performance of QoS in converged Networks environment in which many infrastructure WLANs are deployed in the same geographical area. The following method is use; firstly is to undertake a fundamental investigation to quantify the impact of traffic prioritization on perceived VoIP. Secondly to apply the results to develop efficient traffic prioritization model to benefit end to end voice quality for VoIP applications. Thirdly, to apply the developed models in voice quality monitoring, voice quality optomization (e.g jitter, delay and packet loss optimization) to meet The Mean Opinion Score- Listening Quality (MOS-LQ).

Finally, based on the experiment gained in VoIP simulation testing in converged networks, traffic prioritization modelling can be good tool for predicting the voice and video quality in the VoIP networks. The VoIP simulator models the network and predicts the performance given by the traffic load. Finally, it is learned that traffic priotization gives IP providers a preemptive oppurtunity to maintain quality by increasing bandwidth, performing load balancing and rerouting traffics. This will ensure their ability to deliver consistent and excellent voice quality to users.

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